Lecturer 4, 5

Pulse Code Modulation

Prepared by Prof

Mahmoud Ahmed Attia Ali

Department of Electronics and Communications
Faculty of Engineering
Tanta University
Egypt

TIP
For additional advice see
Dale Carnegie Training®
Presentation Guidelines

Contents

- Generation of PCM Signal.
- Quantization and Coding.
- Compading (Compression & Expansion):
    - μ Law.
    - A Law
- American and European Standard.
- Intersymbol Interference.

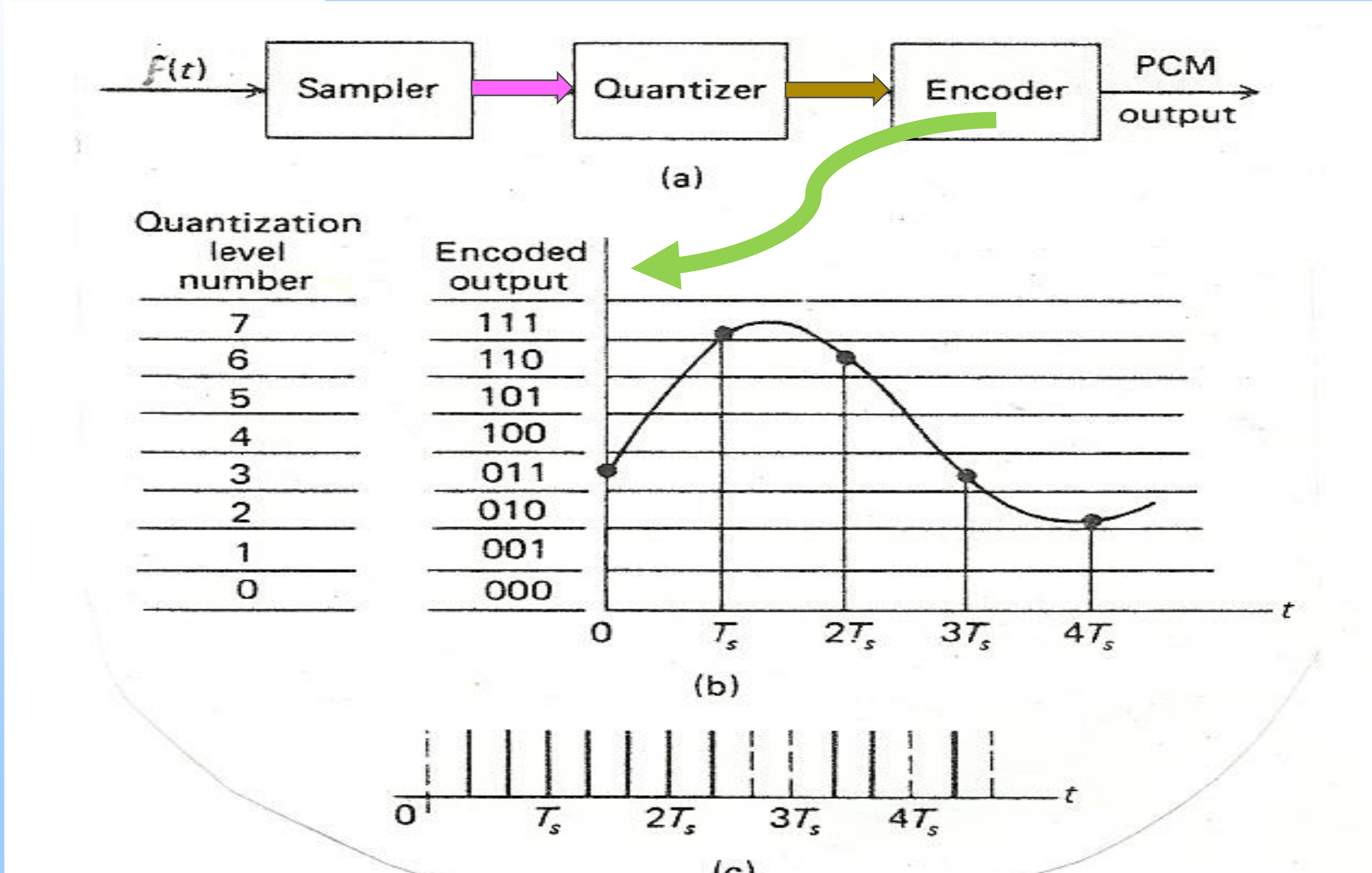
PCM

Generation of PCM

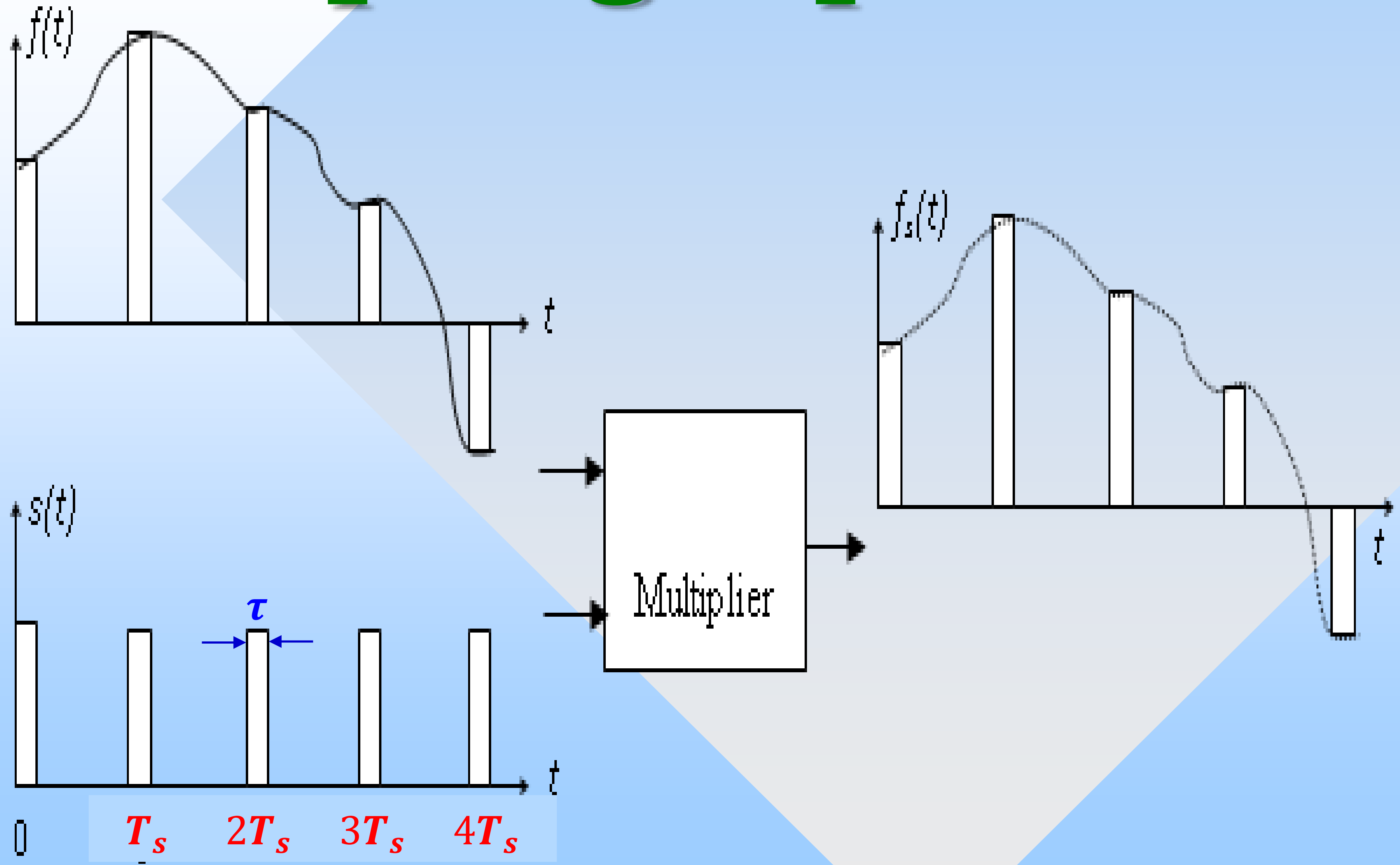
Consists of three processes:

- **Sampler:** Message signal $f(t)$ is first sampled by a rate $f_s > 2f_m$.
- **Quantizer:** Sample values are then quantized to a certain levels.
- **Encoder:** Quantization levels are encoded into binary sequence.

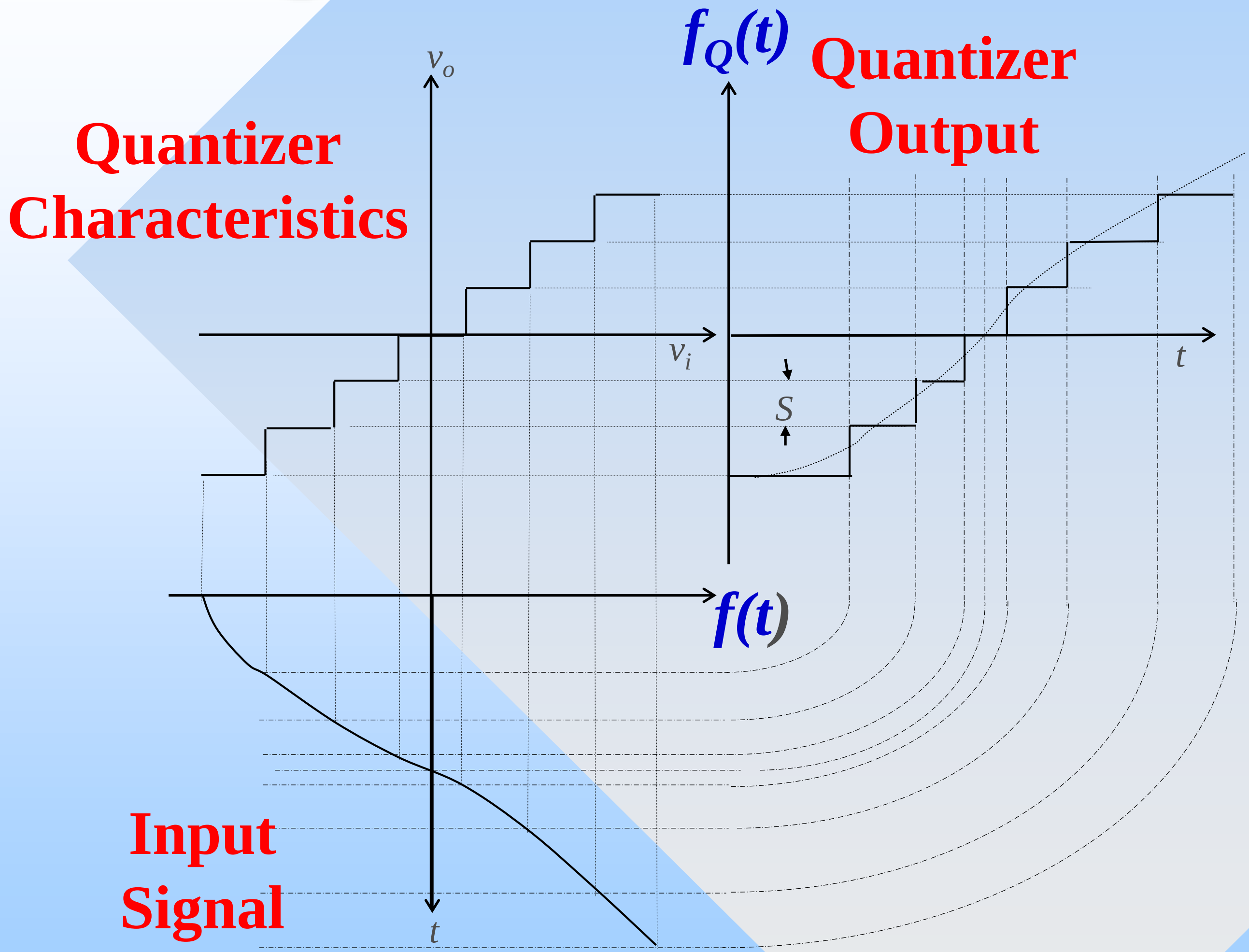
PCM Block Diagram



Sampling Operation



Quantization



Quantization Error

- Difference between original signal $f(t)$ and its quantized approximation $f_Q(t)$
- Why termed as quantization noise?
  - Affects the signal amplitude.
  - May be added to or subtracted from the signal.
  - Expected average value is zero.
- Maximum value is ½ least significant

$$Q_{maximunm} = \frac{v_{LSB}}{2}$$

- Upon reconstruction:
  - Added noise may be removed.
  - Sometimes errors take place.

$\overline{e^2} = \frac{S^2}{12}$

Quantization Error

- Mean square quantization error voltage at quantizer output can be shown to be:

$$\overline{e^2} = S^2/12$$

- Quantization error voltage is: $N_{ov} = \overline{e} = S/\sqrt{12}$
- Signal voltage at the quantizer output is:

$$S_{ov} = SM/2$$

- Signal to quantization error voltage ratio:

$$\frac{S_{ov}}{N_{ov}} = \sqrt{3}\ M$$

- Signal to quantization error power ratio :

$$\frac{S_v}{N_o} = 3\ M^2 = -4.8 + 20\ Log\ M\ dB$$

Quality of Quantization

- Quality of approximation improved by reducing the step size.
- Tests for speech indicate that:
  - 2 levels are understandable but quite noisy.
  - 8 or 16 levels are sufficient for a good intelligibility.
  - 128 or 256 are usually used to ensure high quality.
- Tests for color TV:
  - 64 levels gives only good color TV.
  - 512 levels is used for commercial color TV.

Missed Signal Details

- In quantizing some details are lost.
- It is impossible to reconstruct the original.
- However, there is no need to transmit all signal details:
  - Ears and eyes in hearing and watching is limited:
    - Ear could not distinguish small distortion.
    - Eye has a limited resolution.
  - Also, due to noise, detector will not able to distinguish fine variation.

Probability of Level Error

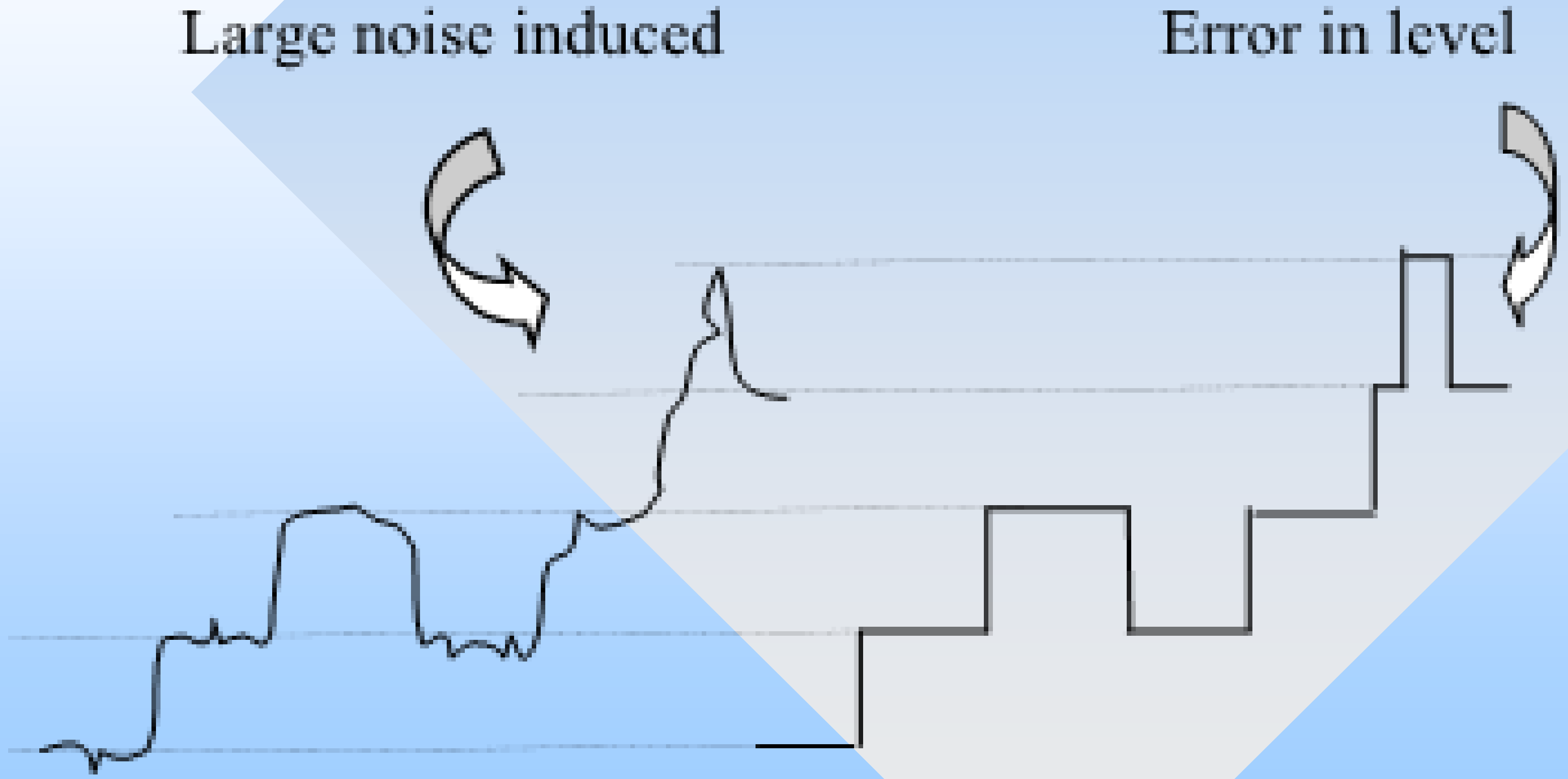


Fig.1.18: Noise Removing or Level Error

Encoding

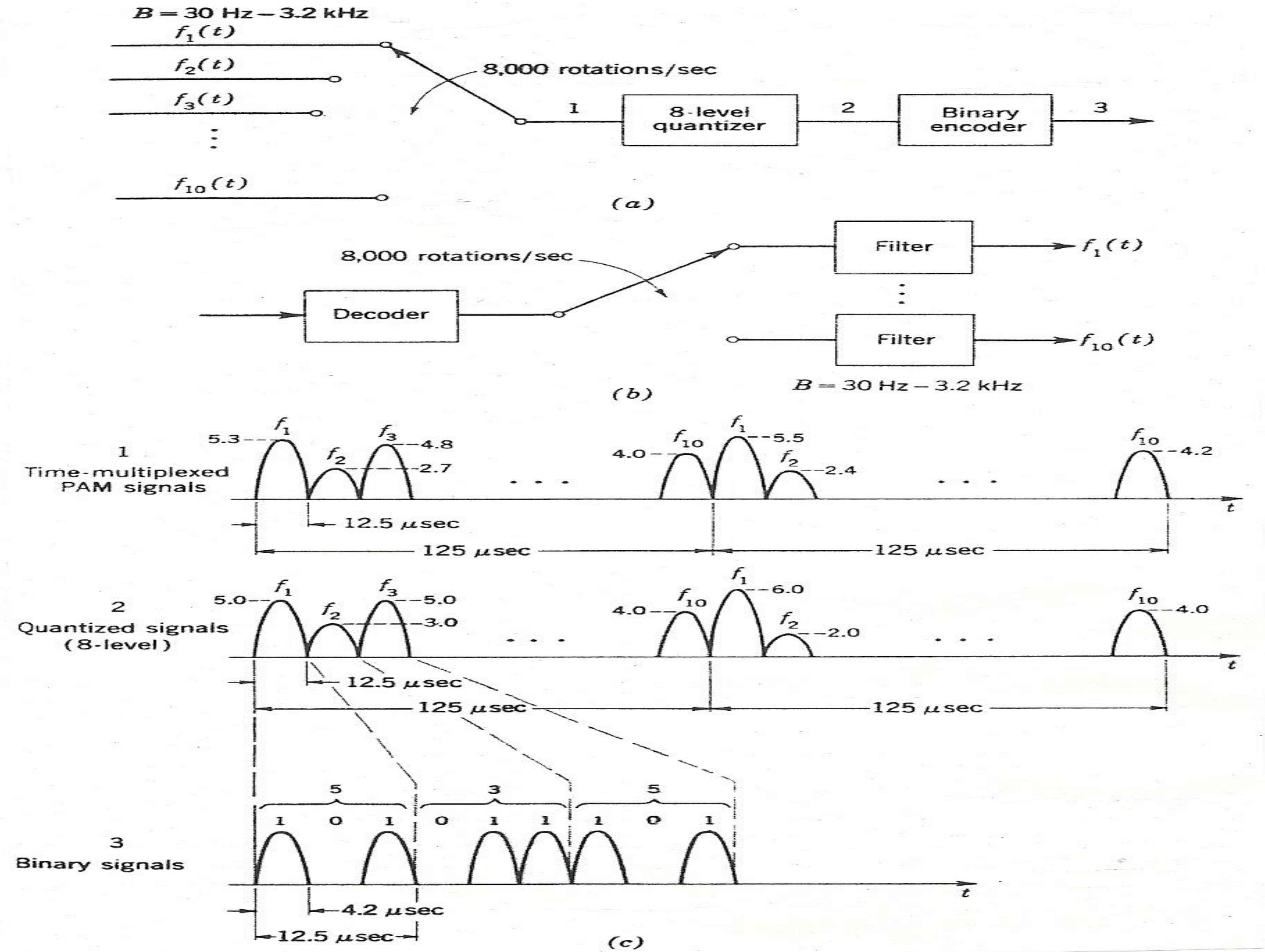
- A binary code where n equals 2.
- Number of quantization levels M is related to the number of bits per sample n as:

$$M = 2^n$$

- Generally, bandwidth for transmission of pulse train is inversely proportional to its width and depends on its shape.
- Therefore, bandwidth is roughly given by the reciprocal of the time slot.

Example

- **Assume 10 channels PCM System**
  - Sampled PAM,
  - Quantized, and
  - PCM using 8 levels quantizer.
- **Required Bandwidths:**
  - PAM signal bandwidth is 1/12.5 μsec = 80 kHz.
  - Quantized PAM signal bandwidth is 80 kHz.
  - PCM signal bandwidth is 1/ 4.2 μsec = 240 kHz.


B = 30 Hz – 3.2 kHz
f1(t)
f2(t)
f3(t)
f10(t)
8,000 rotations/sec
1
8-level quantizer
2
Binary encoder
3
(a)
8,000 rotations/sec
Decoder
Filter
Filter
f1(t)
f10(t)
B = 30 Hz – 3.2 kHz
(b)
1
Time-multiplexed PAM signals
5.3
4.8
2.7
4.0
5.5
2.4
4.2
12.5 μsec
125 μsec
125 μsec
2
Quantized signals (8-level)
5.0
5.0
3.0
4.0
6.0
2.0
4.0
12.5 μsec
125 μsec
125 μsec
3
Binary signals
5
3
5
1 0 1 0 1 1 1 0 1
4.2 μsec
12.5 μsec
t
(c)

Exercises

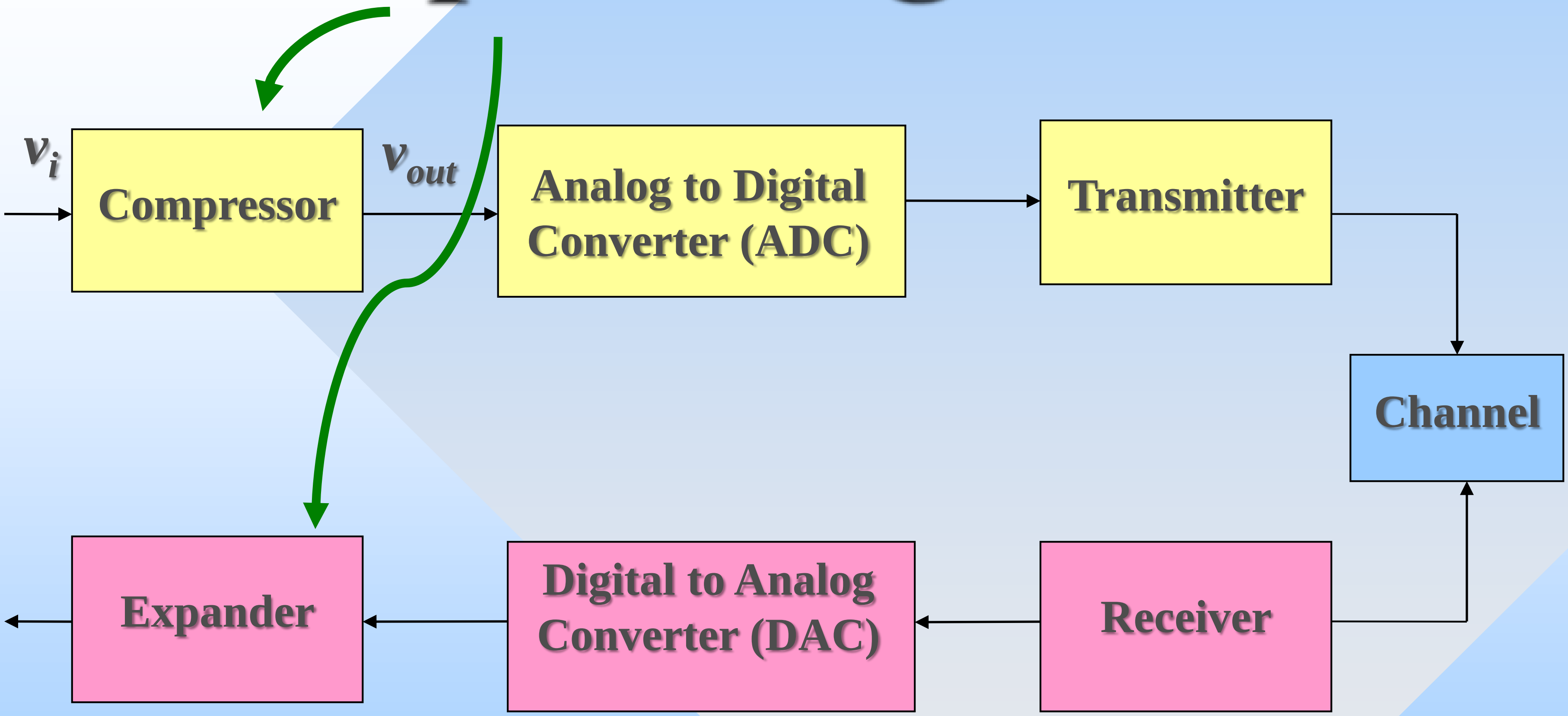
- Indicate the advantages of PCM systems when compared to PAM or Delta techniques.
- What is difference between the unipolar and the bipolar quantization?
- What is difference between mid-rise and mid-tread quantization processes?
- What is difference between the uniform and the nonuniform quantization?

Companding

Reason of Compading

- Small signals will have a poorer signal to quantization noise ratio SQR than large.
- So, it is better to have moderate levels for both low and high variations of the signal:
  - Increase low signal levels to more moderate levels so that SQR could be increased.
  - Reduce high signal levels to more moderate levels in order to decrease high SQR.

Companding Process



Companding Laws

- Companding means compression and expansion.
- Compression by using special designed diodes prior to sampling circuit
- Whereas expansion is attained with diodes after the receiver LPF.
- Voice signal require a constant SQR over a wide dynamic range DR. This requires a logarithmic compression ratio. There are two methods:
  - μ Law Companding.
  - A Law Companding.

μ Law Companding (USA and Japan)

μ Law Compression

$$v_o = \frac{\ln\left[\left\{1 + \mu\left(\frac{v_i}{V_{i,max}}\right)\right\}\right]}{\ln[1+\mu]} V_{o,max}$$

$V_{i,\max}$: Max amplitude of input signal before compression.

$V_{o,\max}$: Max amplitude of output signal after compression.

v_i: Amplitude of input signal before compression for $v_i > 0$

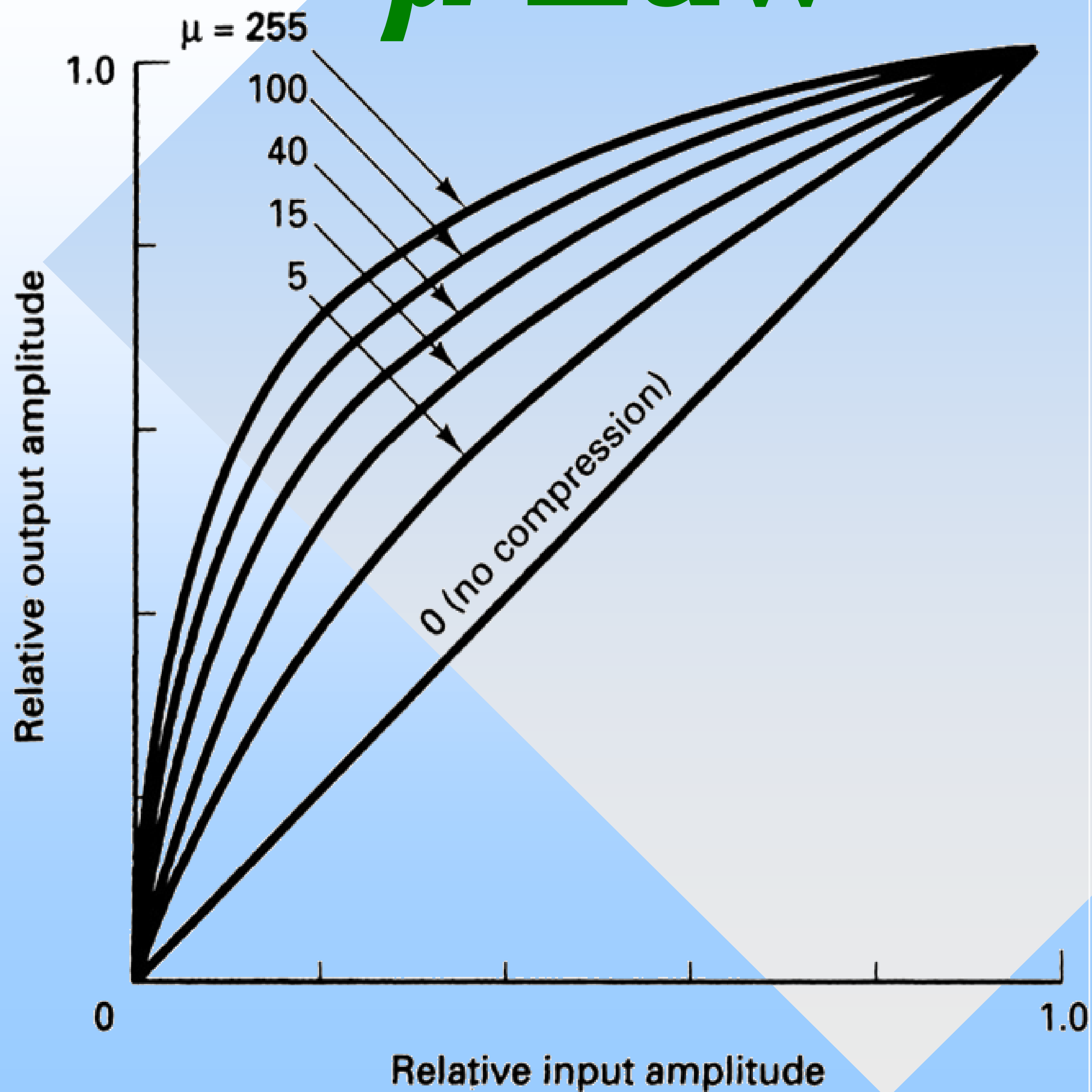
v_o: Amplitude of the output signal after compression.

μ: is a measure for the amount of compression.

μ: determines the range of signal power over which SQR is relatively constant.

- For DR = 40 dB, 7 bit PCM code uses μ = 100.
- For DR = 40 dB, 8 bit PCM code uses μ = 255.

μ Law



μ = 255
100
40
15
5
0 (no compression)
1.0
0
1.0
Relative output amplitude
Relative input amplitude

μ Law Expansion

On reception, received signal will be expanded to satisfy linearity. Henceforth, received signal is:

$$v_{o,r} = \frac{V_{i,r,max}}{\mu}\left[(1+\mu)\left(\frac{v_{i,r}}{V_{o,r,max}}\right) - 1\right] \ , v_{i,r} \geq 0$$

$V_{i,r,\max}$: Max amplitude of input received signal before expansion.

$V_{o,r,\max}$: Max amplitude of output received signal after expansion.

$v_{i,r}$: Amplitude of input received signal before expansion.

$v_{o,r}$: Amplitude of the output received signal after expansion.

A Law Companding (Europe CCITT)

A Law Compression

Compression characteristics is a true logarithm

$$v_o = \frac{\frac{Av_i}{V_{i,max}}}{1+\ln A} V_{o,max}, \quad 0 \le \frac{v_i}{V_{i,max}} \le \frac{1}{A}$$

$$v_o = \frac{1 + \ln\left[\frac{Av_i}{V_{i,max}}\right]}{1 + \ln A} V_{o,max}, \quad \frac{1}{A} \le \frac{v_i}{V_{i,max}} \le 1$$

$V_{i,\max}$: Max amplitude of input signal before compression.

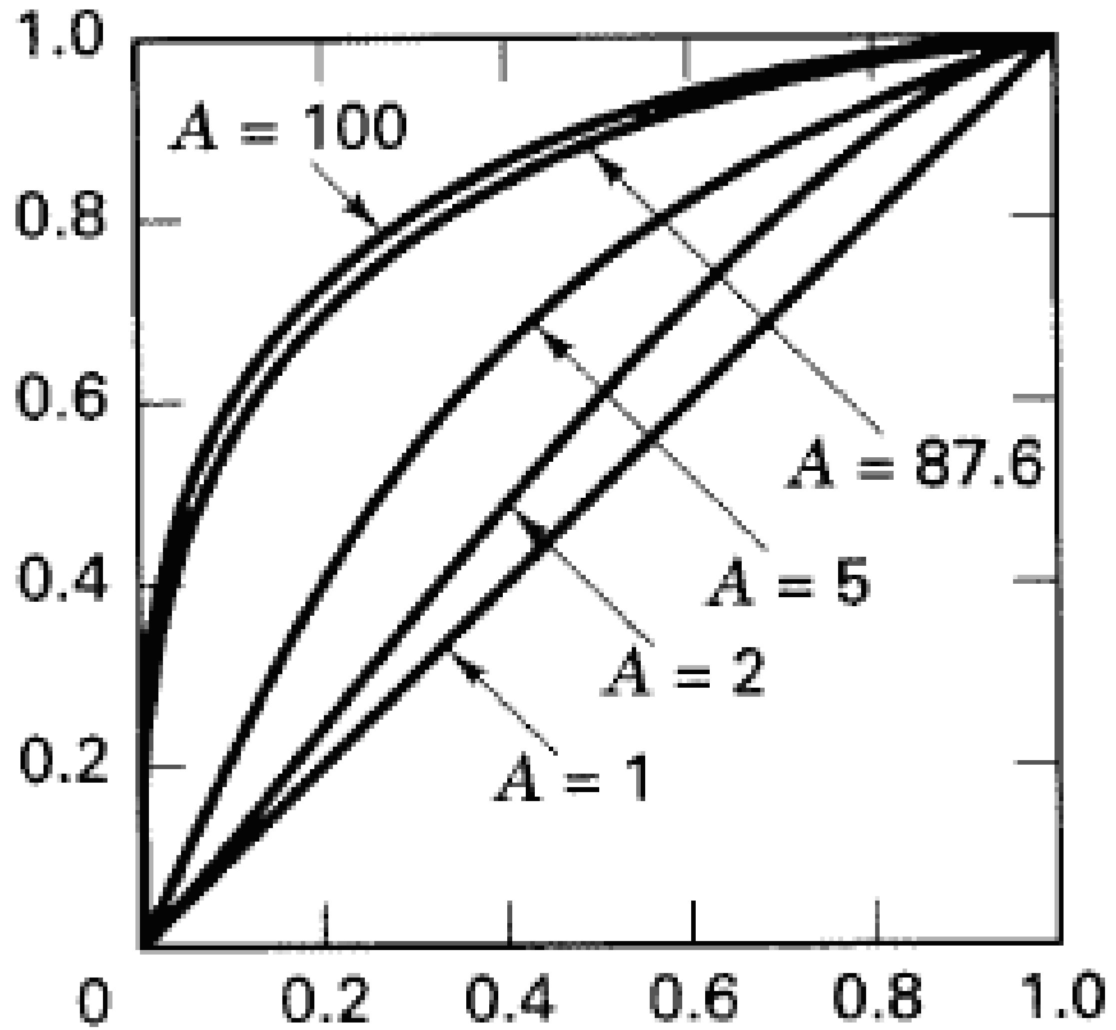
$V_{o,\max}$: Max amplitude of output signal after compression.

v_i: Amplitude of input signal before compression.

v_o: Amplitude of the output signal after compression.

Optimum value for voice transmission is $A = 87.6$.

A Law



Data Rate of PCM

- Speech signal for telephone has $f_m = 4$ kHz.
- So, the sampling rate is $2x4 = 8$ kHz, that is 8000 samples/sec.
- Sampling interval $1/8000 = 125\ \mu$ sec/Frame
- Each sample is encoded into 8 bits/sample.
- So, the data rate for PCM signal is:

$$R_{PCM} = (8000 \text{ samples/sec})(8 \text{ bits/sample})$$
$$= 64 \text{ k bits/sec}$$

North American PCM T1 Standard

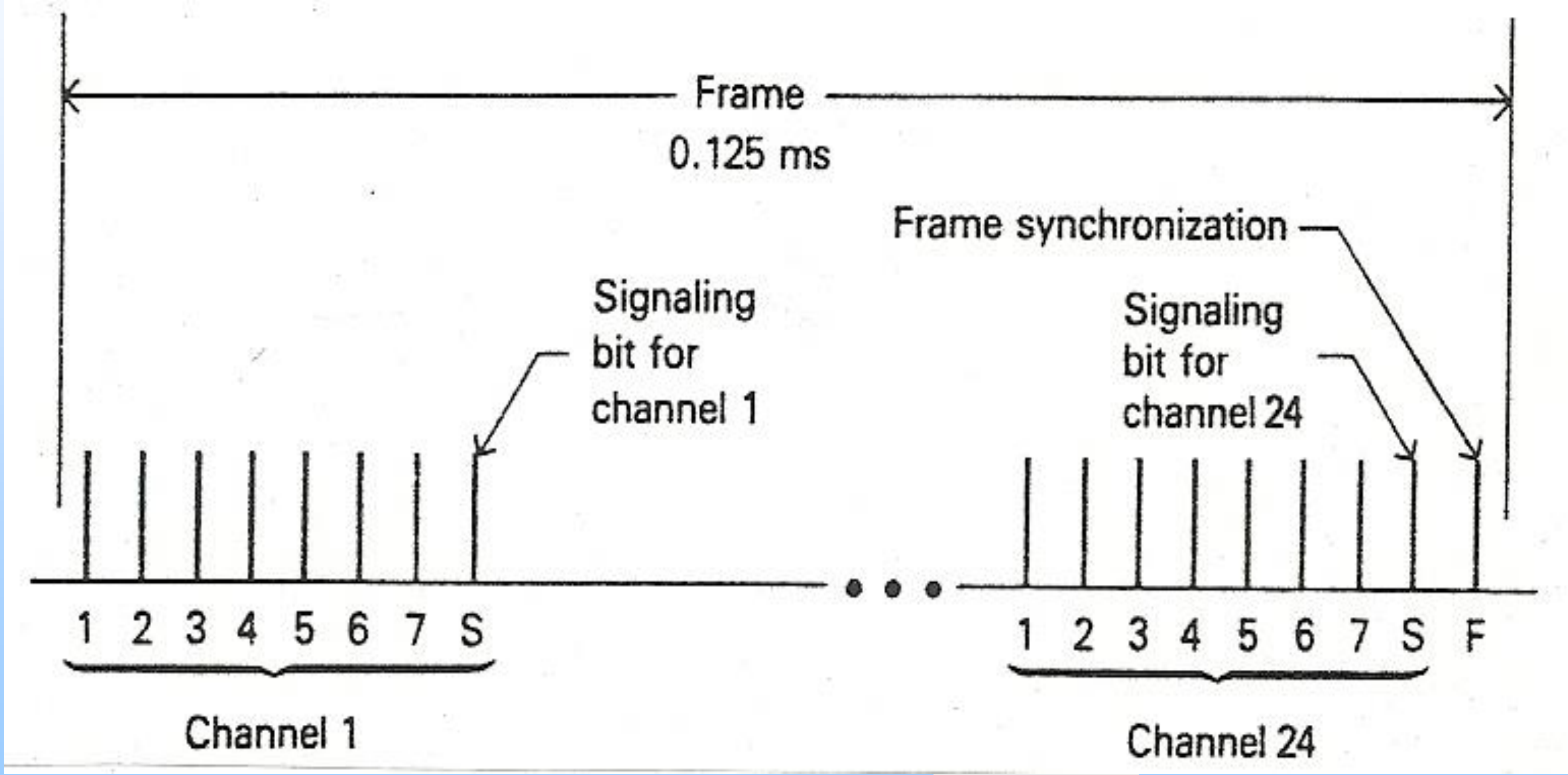
- Bell System in USA introduced 24 channel PCM in1960s for digital voice over short haul distances of 10 to 50 miles “T1”.
- T1 has found widespread adoption in US, Canada, and Japan.

PCM T1 Standard

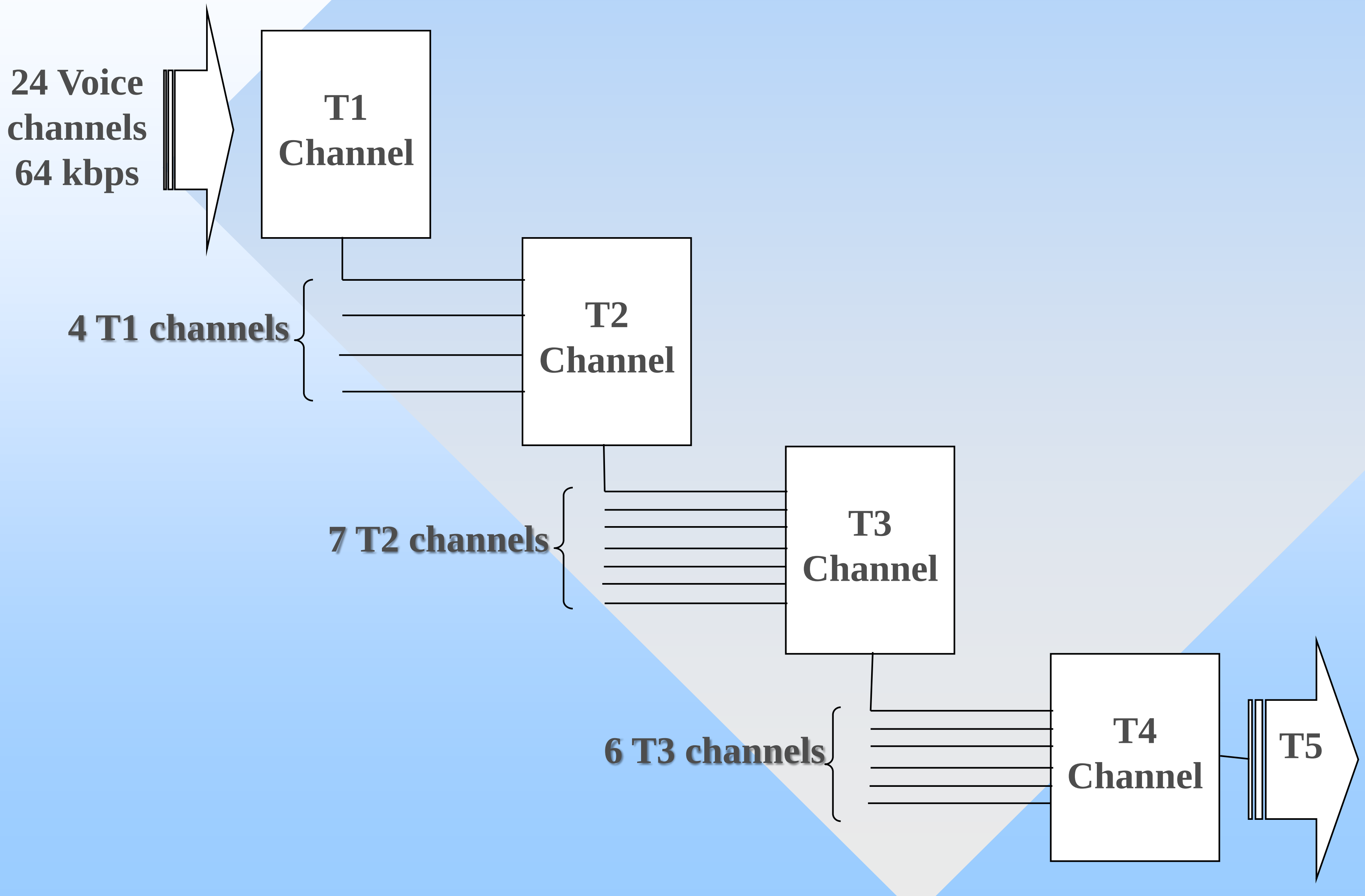
- Early, it uses $2^7 = 128$ quantization levels.
  - Each sample is quantized into 7 bits.
  - 1 bit for establishing calls (signaling).
- Recently, $2^8 = 256$ levels have been adopted for quieter system with less distortion.
- 24 channels are time multiplexed, sampled, and coded into 8 bit PCM formats in addition to 1 bit for frame synchronization.
- The frame consists of $24 \times 8 + 1 = 193$ bits.

$$R_{T1} = (193\ \text{bits/Frame})(1\text{Frame}/125\mu\text{sec}) = 1.544\ \text{Mbps}$$

North American PCM Standard for Short-Haul Telephone [T1 System]



Short to Long Haul PCM Systems



Exercises

Exercise 1.10

- Prove that the transmission data rate of T1 PCM system used in United States, Canada and Japan is given as 1.544 Mbps.

Exercise 1.11

- Estimate the number of channels and the data rate for long haul T2, T3, and T4.

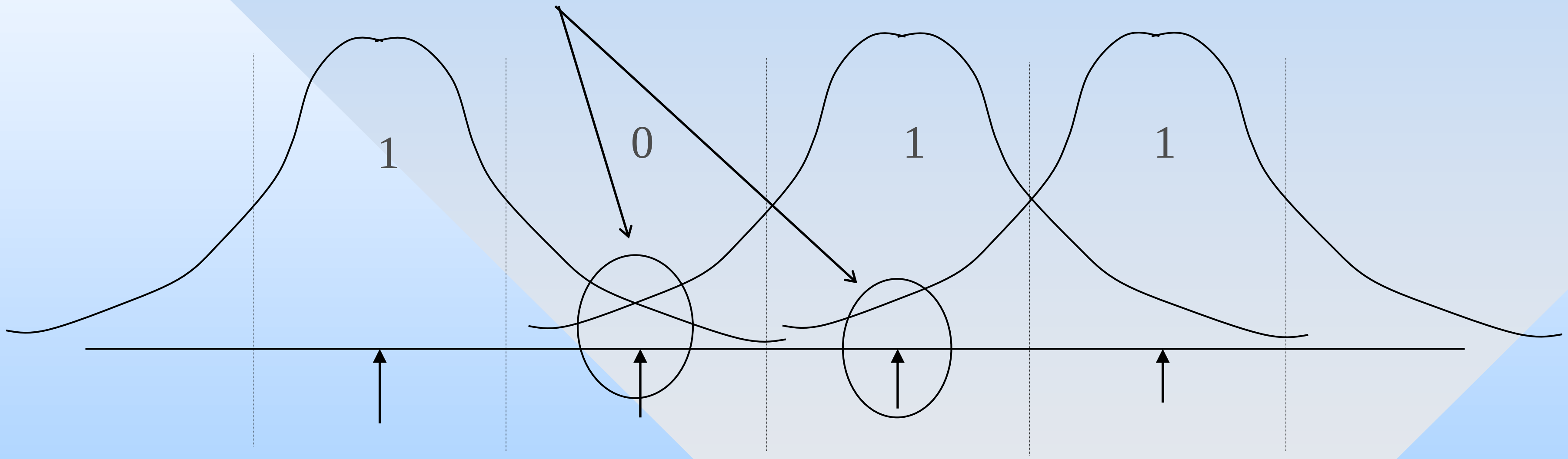
ISI

Intersymbol Interference ‘ISI’

- **First;** Bandwidth of PCM channel is restricted, waveforms will be distorted in a way analogous to crosstalk in PAM.
- **Second;** If digital pulses modulate a carrier for transmission over long distances, pulse shaping is a must.
- This causes pulses to spread out as they transverse channel and overlap into adjacent.

ISI

Inter-symbol Interference



Crosstalk and ISI

- In un-quantized PAM, adjacent time slots are often associated with different message, and crosstalk is appropriate.
- In PCM adjacent bits are generally symbols in the code representation of a single quantized sample, hence the term inter-symbol interference.

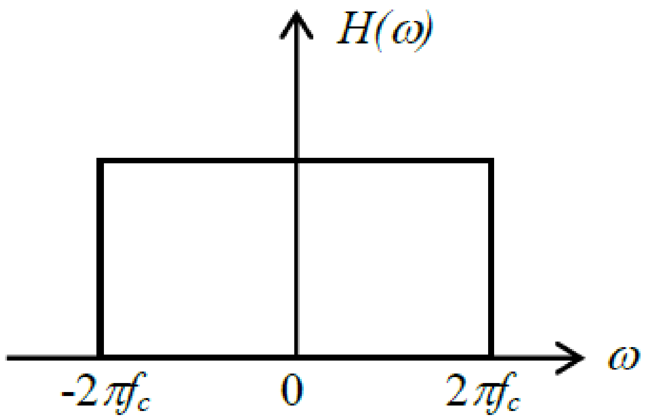
Minimizing ISI

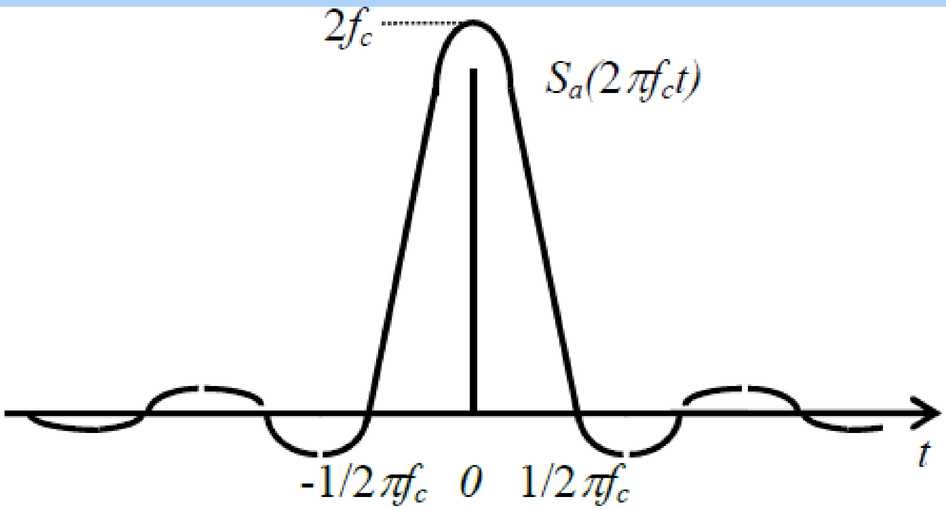
- Widening the transmission bandwidth as much as desired may minimize intersymbol interference.
- Instead, one could seek a way of purposely designing:
    - Signal wave-shapes and
    - Transmission filters used

Sampling Ch[s]

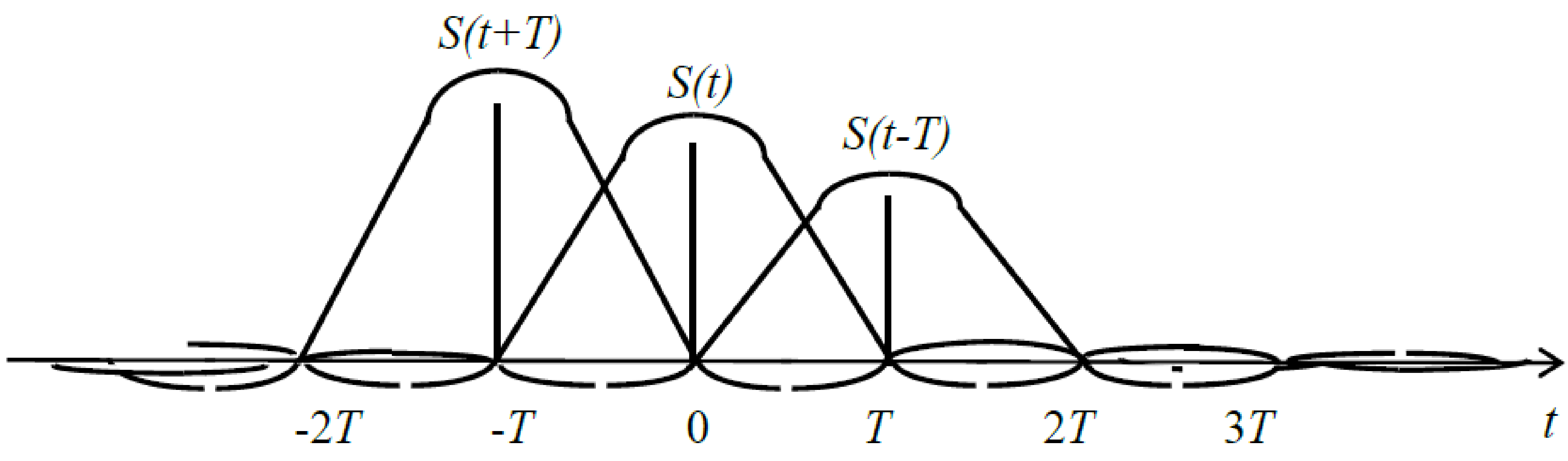
- One wave-shape producing zero ISI is the sampling function: $(sin2\pi f_c t)/2\pi f_c t$
- Is impulse response of ideal LPF at f_c:
- Pulse goes zero at multiples of 1/2πfc
- If sample interval is chosen 1/2πfc adjacent pulses will not interfere

Pulse Providing Zero ISI





Sequence of Digital Pulses with Zero ISI



Practical Difficulties

- It implies the overall ch^s between the transmit and receive is ideal LPF.
- Require precise synchronization. If timing at receiver varies from exact synchronization, zero ISI condition disappears, the tails of all adjacent pulses may add up.
- Also some timing jitter will be present even with most sophisticated synchron.

Raised Cosine

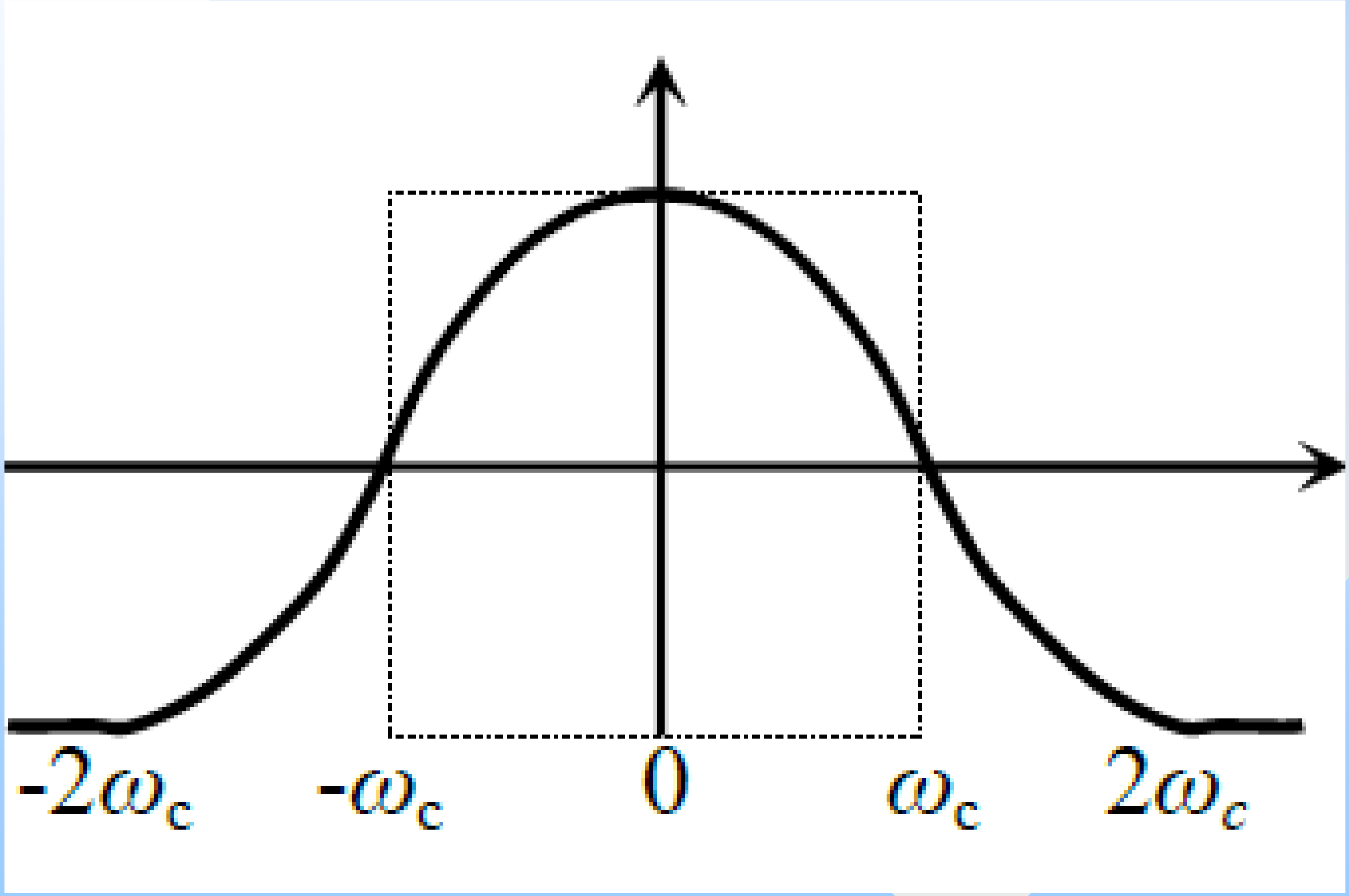
- If pulse has odd symmetry about LPF cutoff, impulse response retains property of zeros at uniformly spaced interval.
- An example is the raised cosine:

$$H(\omega) = \frac{1}{2}\left(1 + \cos\frac{\pi\omega}{2\omega_c}\right) \qquad \& \qquad \omega \le 2\omega_c$$
$$= 0 \qquad\qquad elsewhere$$

- To show symmetry, let $\omega = \omega_c + \Delta\omega$:

$$H(\omega) = \frac{1}{2}\left(1 + \cos\left\{\frac{\pi}{2} + \frac{\pi}{2}\frac{\Delta\omega}{\omega_c}\right\}\right) = \frac{1}{2}\left(1 - \sin\frac{\pi}{2}\frac{\Delta\omega}{\omega_c}\right)$$

Cosine Odd Symmetry



Impulse Response

- The impulse response of a filter with the characteristics shown is given by:

$$h(t) = \frac{\omega_c}{\pi}\frac{\sin\omega_c t}{\omega_c t}\left(\frac{\cos\omega_c t}{1-(2\omega_c t/\pi)^2}\right)$$

- It has the (sin x)/x term multiplied by an additional factor decreases with time.
    - (sin x)/x ensures zero crossing as for LPF.
    - Additional factor reduces the tails of pulses so that it becomes insensitive to timing jitter.

Digital Signaling Formats

Binary Line Coding

- Binary 1's and 0's may be represented in various signaling formats or line codes.
- The most popular are:
  - Unipolar Nonreturn to Zero, UNZ.
  - Unipolar return to Zero, URZ.
  - Bipolar Non return to Zero, PNZ.
  - Bipolar Return to Zero, PRZ.
  - Manchester

Desirable Properties of Line Codes

- •Self-synchronization.
- •Error detection capability:
- •Low probability of error:
- •Suitable spectrum for channel:
- •Transparency:

Self-synchronization

- There is enough timing information built into the code,
- So bit synchronizer can be designed to extract the clock signal.
- So, long series of binary 1’s and 0’s should not cause a problem in time recovery.

Error detection capability

The ability of addition of the channel encoders and decoders

Or incorporating them into the line code.

Low probability of error

- Receivers can be designed to recover data with low probability of error when the input is corrupted by
- Noise or
- Intersymbol interference ISI.

Suitable Spectrum for Channel

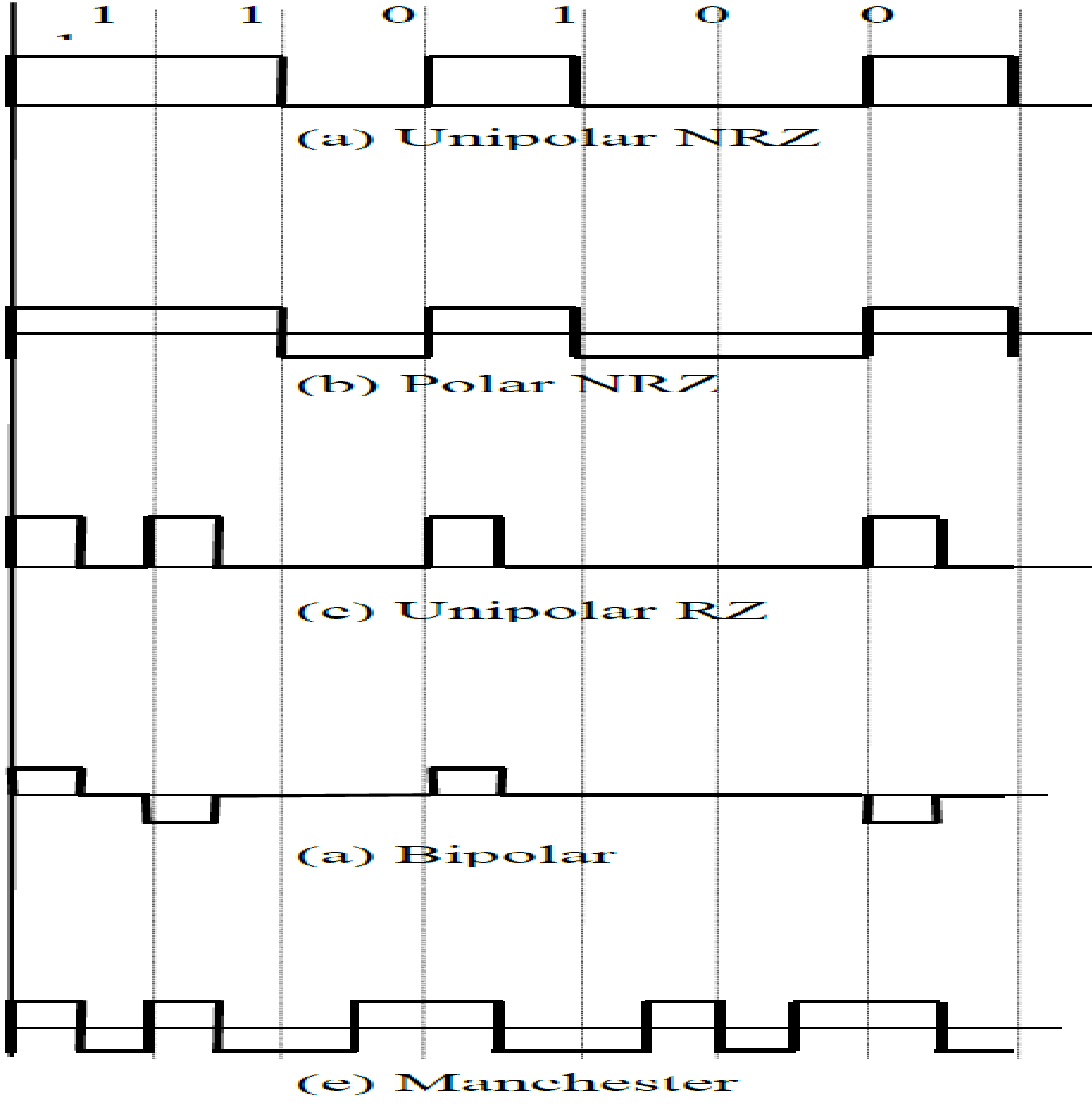
- With ac coupled, the power spectral density of line code should be negligible at frequencies near zero.
- Also, the signal bandwidth needs to be sufficiently small compared to channel bandwidth to minimize ISI.

Transparency

- Every possible sequence of data is faithfully and transparently received.
- A code is not transparent if some sequence will result in a loss of synchronization at the receiver.
- For example, the bipolar format is not transparent since a string of zeros will result in a loss of clocking signal.

Unipolar NRZ

- (on-off keying):
- The binary 1 is represented by a high level and
- a binary 0 by a zero level.
- High level does not return to zero during the binary 1 signaling interval.



1
1
0
1
0
0
(a) Unipolar NRZ
(b) Polar NRZ
(c) Unipolar RZ
(a) Bipolar
(e) Manchester

Polar NRZ

- Simply NRZ:
- Binary 1’s and 0’s are represented by equal
  - » positive
  - » and negative levels.

Unipolar RZ

- The binary 1 is represented by a high level over half of the bit period and then returns to zero.

Bipolar

- **Bipolar:**
- **The binary 1 is represented by alternatively positive or negative values over a half-bit period.**
- **The binary 0 is represented by a zero level.**

Manchester

- Split phase encoding:
- Each binary 1 is represented by a positive half-bit period pulse followed by a negative half-bit period pulse.
- Similarly, a binary 0 is represented by a negative half-bit period followed by a positive half-bit period.

Power Spectra of Line Codes and Comparison

- • Unipolar and Polar Signaling NRZ: The disadvantage of unipolar and polar signaling is the waste of power due to the dc level (Fig.1.36.a and b) so that dc coupled circuits are needed. The advantage is that they are easy to generate (TTL and CMOS

Power Spectra of Line Codes and Comparison

- **Unipolar RZ: In addition to the continuous spectrum, it has discrete spectral lines at the odd multiples of the bit rate. It requires twice of the bandwidth of that of the NRZ codes.**

Power Spectra of Line Codes and Comparison

- Bipolar: There is neither DC nor discrete spectral lines. Its bandwidth is that of NRZ.
- Manchester: There is neither DC nor discrete lines. Its bandwidth is twice that of NRZ.